Cisco Unified SIP Proxy Version 9.0

Product Overview

Cisco® Unified SIP Proxy (USP) is a high-performance, highly scalable Session Initiation Protocol (SIP) proxy server that helps enterprises aggregate their SIP elements into a centralized architecture in order to simplify and improve the flexibility of their network.

The Cisco USP simplifies call routing within multielement SIP networks using call-routing rules to improve control and flexibility of the overall network. For example, an enterprise network may include Cisco Unified Communications Manager for call control, Cisco Unified Border Element for session border control, and Cisco Unified Customer Voice Portal (Cisco Unified CVP) for interactive-voice-response (IVR) as well as other Cisco and third-party SIP-based elements. Cisco USP interconnects these different SIP-based elements so that SIP network design and troubleshooting, when needed, are greatly simplified. Because Cisco USP acts as a “stateless” routing intermediary between these elements, it greatly reduces the call-routing combinations to help identify problems faster and speed troubleshooting. As such, each SIP-based element needs only to route its call activities to Cisco USP to help ensure proper call routing to any other SIP-based element in its network. By forwarding call-routing requests between call-control elements, the Cisco USP provides the means for routing sessions within enterprise and service provider networks.

Cisco Unified SIP Proxy Version 9.0 provides important new features not available in previous versions. Cisco USP Version 9.0 runs in a virtualized OS environment, such as VMware, on the Cisco Unified Computing System™ (Cisco UCS®) Series of servers (refer to Figure 1), and includes the Cisco UCS E-Series Server modules that are installable in Cisco 2900, 3900, 3900E, and the 4000 Series Integrated Services Routers. On the other hand, Cisco USP Versions 8.5.7 and earlier run on the Cisco Integrated Services Routers Generation 2 (ISR G2) Services-Ready Engine (SRE) blade, but are not supported on the Cisco UCS servers, they do not operate in virtualized OS environments, and they do not support all the features that Cisco USP Version 9.0 and later provide.

Figure 1. Cisco UCS Servers (UCS B, C, and E) on Which Cisco USP Version 9.0 Operates
Applications

By simplifying SIP-based call routing, Cisco Unified SIP Proxy Version 9.0 enables a broad range of unified communications applications and services, as described in the following paragraphs.

Cisco Unified Border Element Scalability and Load Balancing
Cisco Unified Border Element (CUBE) is Cisco’s enterprise session border controller providing demarcation, security, interworking, session control, and demarcation in SIP trunking deployments. Cisco Unified SIP Proxy provides a central route point for management of multiple CUBEs, helping simplify and scale large SIP trunking deployments. You can establish logical separations and use a single Cisco Unified SIP Proxy for either or both ingress and egress traffic. You can apply load balancing and rule-based routing, and provide interfacing to the SIP trunk signal normalization where needed (Figure 2).

If a CUBE is unavailable, Cisco Unified SIP Proxy can intelligently reroute to an alternate CUBE. When that CUBE returns to service, Cisco Unified SIP Proxy resumes sending traffic back to that original CUBE. This design enables need-based growth of the service provider interconnect and also avoids risk associated with a single point of failure for the border element.

Figure 2. Cisco Unified Border Element Scalability and Load Balancing in a SIP Trunking Deployment

SIP Trunk for Contact Center
Whether for inbound or outbound traffic, Cisco Unified SIP Proxy enables routing and management across contact center time-division multiplexing (TDM) and SIP trunks. Routing is provided across gateways connecting outside the network as well as across multiple Cisco Unified CVPs. If a Cisco Unified CVP or gateway is unavailable, Cisco Unified SIP Proxy can intelligently reroute to an alternate Cisco Unified CVP or gateway until that one is available again. You can apply load balancing and rule-based routing, and provide interfacing to the SIP trunk signal normalization where needed (Figure 3).
Cisco Unified Communications Manager and Cisco Unified Communications Manager Express SIP Aggregation

Cisco USP can also simplify the network for an enterprise deploying a distributed call-control network using Cisco Unified Communications Manager at large sites and Cisco Unified Communications Manager Express at the branch offices. Management of SIP dial peers across midsize and large Cisco Unified Communications Manager and Cisco Unified Communications Manager Express networks presents a challenge. As opposed to a full mesh, dial peers can be pointed to Cisco Unified SIP Proxy, which provides a central route point. This process also simplifies the addition and removal of new call-processing agents. If a call-processing agent is unavailable, alternate routing and recovery can be provided. You can apply dial normalization and load balancing as needed (Figure 4).
Cisco Unity PBX IP Media Gateway Integration
Cisco Unity® PBX IP Media Gateways (PIMGs) are used to connect time-division multiplexing (TDM)-based private branch exchanges (PBXs) into Cisco Unity voice messaging systems. Placement of Cisco Unified SIP Proxy in front of the Cisco Unity application enables PIMGs to share Cisco Unity ports, in turn enabling scalability of hybrid TDM PBX and IP messaging deployments (Figure 5).

Figure 5. Cisco Unity PBX IP Media Gateway Integration

Service Provider SIP Interconnect Services
For interconnection among service providers, Cisco Unified SIP Proxy enables normalization of dial strings and SIP signaling variants. Cisco Unified SIP Proxy also provides routing and load balancing among SIP elements, including Cisco Unified Border Elements (Figure 6).

Figure 6. Service Provider SIP Interconnect

Product Architecture
Cisco Unified SIP Proxy Call Processing
The Cisco Unified SIP Proxy is a call and dialog stateless SIP proxy, meaning that after Cisco USP determines the correct SIP-based routing, it withdraws from the signaling interaction and allows the SIP-based elements to perform midcall signaling directly between one another. In a network with multiple SIP-based elements, this stateless proxy function greatly simplifies the various SIP protocol interactions between these elements. Furthermore, Cisco USP does not perform any media-handling functions. Instead, session media flows bypass the Cisco USP and go directly to the SIP-based endpoints, which Cisco USP has interacted with in the session signaling process. The Cisco USP can also modify SIP headers (normalization). Routing and normalization are determined based on administrator-configured policies. Policies are selected based on triggers, administrator-configured conditions that are matched based on information in the SIP message.
As SIP messages come into the proxy, a determination is made as to whether any prenormalization policies need to be applied. Following prenormalization, new triggers are used to determine application of routing policies. A further series of triggers provides for further header modifications; for example, postnormalization policies after the routing decision has been made. In cases where policy is not asserted, the proxy provides for pass-through of the SIP message (Figure 7).

**Figure 7.** Cisco Unified SIP Proxy Call-Processing Model

You can apply distinct rules to groups of requests to create independent “virtualized proxies” within a single Cisco Unified SIP Proxy. The rules are highly flexible and scalable to form routing or normalization policies.

**Features**

- Proxy for SIP unified communications signaling
- Signaling support: voice, video, fax, physical terminal line (TTY), modem, caller ID, caller name, updates, transfer, forward, hold, conference, status, message-waiting indicator (MWI), dual-tone multifrequency (DTMF) relay, and SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) (presence)
- Address resolution (Domain Name System [DNS]: Type A and SRV and Type NAPTR)
  - Domain name resolution based on RFC 3263, Locating SIP Servers
- TCP, User Datagram Protocol (UDP), and Transport Layer Security (TLS)
- Standard (RECORD ROUTE ON) mode: This mode keeps a record of the SIP sessions it connects. This record can be very valuable in the SIP-based network to determine if the call-routing policies (dial plan) defined within Cisco USP are working as intended
- Lite (RECORD ROUTE OFF) mode: This mode allows Cisco Unified SIP Proxy to run at a higher SIP request rate than the RECORD ROUTE ON call rate by disabling record-route functions. This mode is typically used after the SIP-based network call routing using Cisco USP has been thoroughly tested. For more details about RECORD ROUTE OFF mode, refer to the “Performance” section
• Deployment options: Cisco USP Versions 8.5.7 and earlier runs natively on an SRE server module installed in the Cisco 2900, 3900 and 3900E, or 4000 Series Integrated Services Routers. Cisco USP Versions 9.0 and later are virtualized and run on Cisco UCS Series Servers, including the Cisco UCS E-Series Server modules, which are also modules on Cisco 2900, 3900 and 3900E, or 4000 Series Integrated Services Routers
  ◦ The server modules impose minimal performance impact on the router, thereby allowing for concurrent router applications

Routing
• Routing based on policy
• Configurable multistep routing policies with route-table lookup
  ◦ Configurable match rules (for example, longest prefix, exact match, and fixed-length match)
  ◦ Configurable keys selected from the SIP request: Remote address, local address, request for Uniform Resource Identifier (URI), P-Asserted-Identity (caller ID), diversion, remote-party ID, To, and From; within these headers Cisco Unified SIP Proxy can select the user, host, port, domain, phone number, URI, carrier codes, and location routing numbers
  ◦ Configurable key modifiers (for example, case insensitivity, ignore plus, ignore display characters, etc.)
  ◦ Numerous routing decisions: Forward to a single route, forward to a route group, reject, and chain to another route policy
• Table-based routing for mapping of requests to destinations
  ◦ Support for a large number of routes in a table (10,000+)
  ◦ Routes populated through command-line interface (CLI) or upload of a route file
• Example routing scenarios:
  ◦ URI-based routing (number and name)
  ◦ Call block between specified sources and destinations, including policy-based transit routing (policy may require certain calls to either avoid or take certain routes)
  ◦ Class of restriction
  ◦ Translation of on-net to off-net dial plans (including public switched telephone network [PSTN] and IP-IP); simplifies network management, eliminating the need to configure translations in each call agent
• Percentage and weight-based routing
  ◦ Load balancing among downstream elements based on preset weight
  ◦ Priority values assignable for routing of selected calls; also enables configuration for least-cost routing
• Time policy routing
  ◦ Time(s) in a day, day(s) in a week, day(s) in a month, and month(s) in a year
• Ability to form downstream elements into a single logical group for load balancing and failover
• SIP element health management and monitoring
  ◦ Rerouting around unavailable SIP elements
  ◦ Ping for service availability and restoration of routing when unavailable SIP element is restored
• Rerouting based on redirect responses (Routing policy and postnormalization applies to the new destination specified in the contact header of the redirect response, and provides for sequential forking)
• Transport protocol conversion: TCP, UDP, TLS (For example, an incoming call received over UDP can be forwarded to a destination over TLS)
• Configurable record routing (on/off)
• Global unique caller ID pass-through

**Normalization**

• Normalization of SIP headers based on configurable policy
  ◦ Ability to add, remove, or update headers and header parameters
  ◦ Ability to update URI components such as user, domain, and host and ability to add, remove, and update URI parameters
  ◦ Digit manipulation
  ◦ Address manipulation
  ◦ TEL URI <=> SIP URI conversion
  ◦ Domain conversions
  ◦ Regular-expression processing
• Construction of multistep normalization policies
• Pre- and postnormalization
  ◦ Prenormalization prior to proxy application of routing rules (for example, applied to message coming into the proxy)
  ◦ Postnormalization after proxy application of routing rules (for example, applied to message going out from the proxy)

**Rules-Based Selection of Routing and Normalization Policies**

• Rich set of configurable rules
  ◦ SIP message type (for example, request and response)
  ◦ SIP method: INVITE, UPDATE, REFER, PRACK, BYE, SUBSCRIBE, NOTIFY, unsolicited NOTIFY, MESSAGE, PUBLISH, REGISTER, INFO, OPTIONS, and any custom or future SIP extensions
  ◦ Request-URI: User, host, phone number, etc.
  ◦ Local and remote IP, port, and protocol of the received SIP message
  ◦ Network name of the incoming and outgoing request (A network is a set of SIP listening points.)
  ◦ Transport protocol
  ◦ Regular expression match on any SIP header
  ◦ Time policy check
  ◦ SIP response code
  ◦ Mid-dialog message check
• Call Admission Control
  ◦ Call-counting based
Security and Privacy

- TLS (bidirectional)
- Through-header stripping (for topology hiding)
- User privacy (RFC 3325 P-Asserted ID: Removes P-Asserted ID when receiving a message from an element not configured as trusted, and removes P-Asserted ID and Privacy header when forwarding a message to an element not configured as trusted)

Network Design

- Multiple IP addresses (up to eight) to provide for flexible configuration and network topology design; you can group IP addresses to form networks and apply rules on these networks
- Multiple SIP listening points; each listen point can have a configurable port
- “Virtualized proxies” with multiple independent routing and normalization processing in a single server
- Redundancy through clustered network design for high availability
  - Clusters addressed as Fully Qualified Domain Names (FQDNs); DNS resolution through Service (SRV) record
  - Virtual IP addressing using Hot Standby Router Protocol (HSRP)
- Very high scalability with clustering of multiple Cisco Unified SIP Proxies
  - Hierarchical and peer requests among clustered Cisco Unified SIP Proxies, either as server-based or virtualized platforms

Management

- Flexible management through GUI and CLI
- Monitoring of system status using Simple Network Management Protocol (SNMP) MIBs
- Load of preexisting configurations onto the Cisco Unified SIP Proxy module
- Copy of configurations off the Cisco Unified SIP Proxy module
- Graceful shutdown and restore, allowing for completion of transactions in process
- RADIUS accounting for SIP events
- SIP message logging for call monitoring
- Trace logging for troubleshooting
- FTP access to Cisco Unified SIP Proxy for easy download of trace logs, SIP message logs, configuration files, and route files, and upload of configuration files and route files
- SIP message metrics logging (peg counting); for example, count of incoming and outgoing messages over a period of time and logging to a file
- Detailed call statistics with call attempt, success, and failure rates per element
- Database store for debs and logs
  - Selectively log messages using regular expressions
  - Search through stored log messages
  - Log up to 1 million log messages
**Supported Standards as a SIP Proxy**

- IETF RFC 2246: TLS Protocol Version 1.0
- IETF RFC 2327 SDP: Session Description Protocol
- IETF RFC 2617 HTTP Authentication: Basic and Digest Access Authentication
- IETF RFC 2782: A DNS record route (RR) for specifying the location of services (DNS SRV)
- IETF RFC 2806: URLs for Telephone Calls
- IETF RFC 2976: The SIP INFO Method
- IETF RFC 3204: MIME media types for ISUP and QSIG Objects
- IETF RFC 3261 SIP: Session Initiation Protocol
- IETF RFC 3262: Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- IETF RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers
- IETF RFC 3264: An Offer/Answer Model with the Session Description Protocol (SDP)
- IETF RFC 3265: Session Initiation Protocol (SIP)-Specific Event Notification
- IETF RFC 3311: The Session Initiation Protocol (SIP) UPDATE Method
- IETF RFC 3325: Private Extensions to the SIP for Asserted Identity within Trusted Networks
- IETF RFC 3326: The Reason Header Field for the Session Initiation Protocol (SIP)
- IETF RFC 3515: The Session Initiation Protocol (SIP) Refer Method
- IETF RFC 3665: Session Initiation Protocol (SIP) Basic Call Flow Examples
- IETF RFC 3666: Session Initiation Protocol (SIP) Public Switched Telephone Network (PSTN) Call Flows
- IETF RFC 3725: Best Current Practices for Third-Party Call Control (3PCC) in the SIP
- IETF RFC 3856: A Presence Event Package for the Session Initiation Protocol (SIP)
- IETF RFC 3891: The Session Initiation Protocol (SIP) "Replaces" Header
- IETF RFC 3892: The Session Initiation Protocol (SIP) Referred-By Mechanism
- IETF RFC 4480 RPID: Rich Presence Extensions to the Presence Information Data Format (PIDF)
- SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)

**Ordering**

The Cisco Unified SIP Proxy (USP) Version 9.0 introduces a major enhancement in the ordering and licensing processes compared to prior versions. This major enhancement is the introduction of Cisco Smart Licensing as the mechanism for ordering, enabling, and tracking the software licensing associated with Cisco USP.

Cisco Smart Licensing is a licensing innovation that is being applied to a wide range of Cisco software products, both those based on Cisco IOS® Software and those that aren’t, in order to achieve a greater level of licensing consistency and improve the overall user experience for Cisco customers. For more information about Cisco Smart Licensing, please visit: [http://www.cisco.com/c/en/us/products/abt_sw.html](http://www.cisco.com/c/en/us/products/abt_sw.html).
With the introduction of Cisco Smart Licensing as part of Cisco USP Version 9.0, the ordering for Cisco USP is greatly simplified. You need only three Cisco USP Smart Licensing product IDs (PIDs), as shown in Table 1.

Table 1. Cisco Unified SIP Proxy Version 9.0 Software Licensing Product IDs

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Description</th>
<th>Minimum Version of Cisco Unified SIP Proxy Software Required</th>
</tr>
</thead>
<tbody>
<tr>
<td>L-CUSP-SL-100</td>
<td>Cisco Unified SIP Proxy version 9.0 licensed for 100 SIP Requests per second (CPS) processing</td>
<td>9.0.0</td>
</tr>
<tr>
<td>L-CUSP-SL-30</td>
<td>Cisco Unified SIP Proxy version 9.0 licensed for 30 SIP Requests per second (CPS) processing</td>
<td>9.0.0</td>
</tr>
<tr>
<td>L-CUSP-SL-10</td>
<td>Cisco Unified SIP Proxy version 9.0 licensed for 10 SIP Requests per second (CPS) processing</td>
<td>9.0.0</td>
</tr>
</tbody>
</table>

Some of the beneficial aspects of these Cisco USP Smart Licensing PIDs include:

- They are transferable: These licenses can be moved between Cisco USP Version 9.0 virtual platforms as needed to support your SIP network deployment model. Furthermore, if these licenses are initially installed on a Cisco UCS E-Series virtualized server platform, they can be transferred to a Cisco UCS B-Series virtualized server platform at any time without any license upgrade or transfer fee.

- They are additive: You don’t need to purchase confusing licensing upgrade PIDs, as required in prior versions. If you initially order the 100 SIP Requests per second (RPS) Cisco USP Smart Licensing PID, but eventually need a higher level of CPS licensing, such as 200 CPS, then you can order an additional 100 CPS Smart Licensing PIDs, for a total of 200 CPS Cisco USP Licenses.

- They are incremental: You can increase CPS scale by increments of 10 CPS on any Cisco USP Version 9.0 virtual platform up to the full operating capacity of the platform.

In summary, the Cisco USP Smart Licensing PIDs provide dramatic improvements in the flexibility of Cisco USP licensing that have not been possible previously.

If you are purchasing Cisco USP for the first time, Cisco recommends that you acquire and deploy the new virtualized Cisco USP Version 9.0, because this version (and future Cisco USP versions) is where new Cisco USP features will be added. Previous Cisco USP versions running natively on the Cisco Services-Ready Engine (SRE) (described in the next section) will be supported with respect to fixing product defects (bug fixes) only, and no new feature enhancements will be made.

Ordering for Existing Customers with Versions Earlier than 9.0

Cisco will continue to support Cisco USP versions prior to 9.0 running natively on the Cisco SRE, but only for fixing product defects, not for any new feature enhancements. Therefore, you will need to transition to Cisco USP Version 9.0 to gain the newest features and functions. If you are an existing Cisco USP user and transition your installations by September 2016, Cisco will provide the replacement Cisco USP licenses (based on Smart Licensing, as described previously) at no charge. If you are unable to transition by September 2016, Cisco reserves the right to charge for the new Cisco USP Version 9.0 Smart Licensing as per the terms of existing Cisco USP licensing.

User self-server migration of existing Cisco USP 8.0 SWIFT licenses to Cisco USP 9.0 Smart Licenses will be supported in the Cisco Smart Licensing Portal after June 2015. Existing users who have an urgency to migrate the licenses prior to June 2015 can contact the Cisco USP Product Management team @ cusp-license-migration@cisco.com with relevant information.
Cisco Software Support Service
For all customers who either purchase Cisco USP Version 9.0 as a new user or transition to Cisco USP Version 9.0 from a previous version, Cisco will require the purchase of Cisco Software Support Service (SWSS) contracts to receive both technical support and new version enhancements. Information regarding SWSS contracts for Cisco USP v9.0 will be posted by March 31, 2015, at the SWSS partner and customer websites, as follows:

- SWSS Partner website
- SWSS Customer website
- SWSS Disti
- SWSS Support Guide

Ordering Cisco Unified SIP Proxy with Cisco Services-Ready Engine
Although Cisco Unified SIP Proxy will remain available on the SRE blade as a nonvirtualized software license, it will not include support for Cisco Unified SIP Proxy Version 9.0 or later. If you want ongoing software enhancements for Cisco USP, you must transition to Cisco USP Version 9.0 running on the Cisco UCS server platforms.

Cisco Unified SIP Proxy on the SRE is supported on specific Cisco ISR platforms, as explained in Tables 2 and 3.

Table 2. Supported ISR Platforms for Cisco Unified SIP Proxy Version 8.5.0 and Earlier on SRE Blade for ISR G2 Routers

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Description</th>
<th>Minimum Version of Cisco Unified SIP Proxy Software Required</th>
</tr>
</thead>
<tbody>
<tr>
<td>SM-SRE-700-K9</td>
<td>Cisco Service Ready Engine 700 Service Module</td>
<td>8.5.1</td>
</tr>
<tr>
<td>SM-SRE-900-K9</td>
<td>Cisco Service Ready Engine 900 Service Module</td>
<td>8.5.1</td>
</tr>
</tbody>
</table>

Table 3. Cisco Unified SIP Proxy Version 8.5.0 and Earlier ISR Platform Support for SRE Module

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Cisco ISR Platform</th>
<th>Minimum Cisco IOS Software version on ISR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Cisco 2911 and 2921</td>
<td>15.0(1)M</td>
</tr>
<tr>
<td></td>
<td>Cisco 2951 and 3900</td>
<td>15.1(1)T</td>
</tr>
<tr>
<td></td>
<td>Cisco 3900E</td>
<td>Not applicable</td>
</tr>
<tr>
<td></td>
<td>Cisco 4000</td>
<td></td>
</tr>
<tr>
<td>SM-SRE-700-K9</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>SM-SRE-900-K9</td>
<td>Yes</td>
<td>Yes</td>
</tr>
</tbody>
</table>

Cisco Unified SIP Proxy on the SRE platform does not use Cisco Smart Licensing, but employs a counted feature license based on the maximum number of new incoming SIP requests per second. Requests that belong to an existing dialog, including SIP responses, are not counted. Refer to Table 4 for Cisco Unified SIP Proxy SIP requests per second supported on the SRE modules.

Table 4. Cisco Unified SIP Proxy Licenses Supported on SRE Modules

<table>
<thead>
<tr>
<th>SIP Requests per Second</th>
<th>SM-SRE-700-K9</th>
<th>SM-SRE-900-K9</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>10</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>30</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>100</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>200</td>
<td>No</td>
<td>Yes</td>
</tr>
</tbody>
</table>
Refer to Tables 5, 6, and 7 for the Cisco Unified SIP Proxy feature licenses available for the SRE modules.

**Table 5.** Cisco Unified SIP Proxy Feature Licenses - Used for Preinstall on SRE Shipments from the Factory

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FL-CUSP-2</td>
<td>Cisco USP Feature License for 2 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-10</td>
<td>Cisco USP Feature License for 10 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-30</td>
<td>Cisco USP Feature License for 30 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-100</td>
<td>Cisco USP Feature License for 100 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-200</td>
<td>Cisco USP Feature License for 200 SIP requests/second</td>
</tr>
</tbody>
</table>

**Note:** The licensed number of requests/second refers to new incoming SIP requests. Requests that belong to an existing dialog, including SIP responses, are not counted.

**Note:** Product with part number FL-CUSP-200 is supported on product with part number SM-SRE-900-K9 only.

A Cisco Unified SIP Proxy license can be added to an SRE (those with part number SM-SRE-700-K9 or SM-SRE-900-K9) after shipment from the factory. In this case, a spare feature license is used. You can receive these spare licenses through standard delivery or in an email message.

- Refer to Table 6 for Cisco Unified SIP Proxy spare feature licenses that you want to receive by standard delivery.
- Refer to Table 7 for Cisco Unified SIP Proxy feature licenses that you want to receive through an email message (e-delivery).

**Table 6.** Cisco Unified SIP Proxy Spare Licenses for SRE Modules - Standard Delivery

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FL-CUSP-2</td>
<td>Cisco USP Feature License for 2 SIP requests/second Spare</td>
</tr>
<tr>
<td>FL-CUSP-10</td>
<td>Cisco USP Feature License for 10 SIP requests/second Spare</td>
</tr>
<tr>
<td>FL-CUSP-30</td>
<td>Cisco USP Feature License for 30 SIP requests/second Spare</td>
</tr>
<tr>
<td>FL-CUSP-100</td>
<td>Cisco USP Feature License for 100 SIP requests/second Spare</td>
</tr>
<tr>
<td>FL-CUSP-200</td>
<td>Cisco USP Feature License for 200 SIP requests/second Spare</td>
</tr>
</tbody>
</table>

**Note:** Product with part number FL-CUSP-200= is supported on product with part number SM-SRE-900-K9 only.

**Table 7.** Cisco Unified SIP Proxy Spare E-delivery Licenses for SRE Modules

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Description</th>
</tr>
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<tbody>
<tr>
<td>L-FL-CUSP-2</td>
<td>Cisco USP Feature License for 2 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-CUSP-10</td>
<td>Cisco USP Feature License for 10 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-CUSP-30</td>
<td>Cisco USP Feature License for 30 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-CUSP-100</td>
<td>Cisco USP Feature License for 100 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-CUSP-200</td>
<td>Cisco USP Feature License for 200 SIP requests/second (e-delivery)</td>
</tr>
</tbody>
</table>

**Note:** Part number L-FL-CUSP-200= is supported on part number SM-SRE-900-K9 only.
Cisco Unified SIP Proxy supports upgrade from one license level to a higher one according to the capacity of the module. Refer to Table 8 for Cisco Unified SIP Proxy upgrade licenses. In the case of a multilevel upgrade, for example, to upgrade from 2 SIP requests per second to 100 SIP requests per second, you should order licenses with part numbers FL-CUSP-2U10= and FL-CUSP-10U100=.

Table 8. Cisco Unified SIP Proxy Upgrade Licenses

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>FL-CUSP-2U10=</td>
<td>Cisco USP Upgrade License for 2 to 10 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-10U30=</td>
<td>Cisco USP Upgrade License for 10 to 30 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-10U100=</td>
<td>Cisco USP Upgrade License for 10 to 100 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-30U100=</td>
<td>Cisco USP Upgrade License for 30 to 100 SIP requests/second</td>
</tr>
<tr>
<td>FL-CUSP-100U200=</td>
<td>Cisco USP Upgrade License for 100 to 200 SIP requests/second</td>
</tr>
</tbody>
</table>

Note: Product with part number FL-CUSP-100U200= is supported on product with part number SM-SRE-900-K9 only.

You can receive Cisco Unified SIP Proxy upgrade licenses in an email message. In order to receive upgrade licenses by e-delivery, you must choose an e-delivery type license. Refer to Table 9 for Cisco Unified SIP Proxy e-delivery upgrade licenses.

Table 9. Cisco Unified SIP Proxy Upgrade E-Delivery Licenses

<table>
<thead>
<tr>
<th>Part Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>L-FL-CUSP-2U10=</td>
<td>Cisco USP Upgrade License for 2 to 10 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-CUSP-10U30=</td>
<td>Cisco USP Upgrade License for 10 to 30 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-CUSP-10U100=</td>
<td>Cisco USP Upgrade License for 10 to 100 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-CUSP-30U100=</td>
<td>Cisco USP Upgrade License for 30 to 100 SIP requests/second (e-delivery)</td>
</tr>
<tr>
<td>L-FL-Cisco USP-100U200=</td>
<td>Cisco USP Upgrade License for 100 to 200 SIP requests/second (e-delivery)</td>
</tr>
</tbody>
</table>

Note: Product with part number L-FL-CUSP-200= is supported on product with part number SM-SRE-900-K9 only.

Performance

Performance is limited by both the number of incoming SIP requests specified in the feature license and module processing capability. With Cisco Unified SIP Proxy Versions 8.5 and later, you can operate Cisco Unified SIP Proxy in standard (Record Route On) and lite (Record Route Off) modes.

Standard mode (also referred to as RECORD ROUTE ON) supports the SIP requests-per-second performance described by the feature license installed. For example, the Cisco USP Smart license (part number L-CUSP-SL-100=) authorizes 100 SIP requests per second. This standard mode is used in most deployments because it provides a complete history of SIP requests.

Lite mode (also referred to as RECORD ROUTE OFF) enables Cisco Unified SIP Proxy to run at a higher SIP requests-per-second rate when the record route feature is disabled. In this mode, Cisco USP will authorize a higher number of SIP requests per second. The licensing multiplier between Cisco USP licensing for Standard mode as compared to Lite mode is 2.5x. So, the same Cisco Smart License (part number L-CUSP-SL-100=) running in Lite mode would authorize up to 250 SIP requests per second.
The licensing multiplier, as explained previously, is the same for Cisco USP Smart Licensing and for Cisco USP PAK Licensing. It is important that the Cisco USP Licensing be properly matched to the platform performance in terms of SIP requests per second. The following tables show the difference in performance between Cisco USP 9.0 running on the Cisco UCS E-Series platform as compared to earlier versions of Cisco USP running on the SRE platform.

Table 10 describes the Cisco Unified SIP Proxy maximum performance on the SRE and Cisco UCS E-Series modules in Standard mode (RECORD ROUTE ON).

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Cisco USP 8.5.7 and Earlier on SRE</th>
<th>Cisco USP 9.0 and Later on Cisco UCS E-Series</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>UDP</td>
<td>TCP</td>
</tr>
<tr>
<td>Call routing only</td>
<td>200</td>
<td>140</td>
</tr>
<tr>
<td>Call routing with normalization</td>
<td>140</td>
<td>125</td>
</tr>
<tr>
<td>CVP</td>
<td>250</td>
<td>225</td>
</tr>
</tbody>
</table>

Table 11 describes the Cisco Unified SIP Proxy maximum performance on the SRE and Cisco UCS E-Series modules in Lite mode (RECORD ROUTE OFF).

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Cisco USP 8.5.7 and Earlier on SRE</th>
<th>Cisco USP 9.0 and Later on Cisco UCS E-Series</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>UDP</td>
<td>TCP</td>
</tr>
<tr>
<td>Call routing only</td>
<td>500</td>
<td>350</td>
</tr>
<tr>
<td>Call routing with normalization</td>
<td>350</td>
<td>310</td>
</tr>
<tr>
<td>CVP</td>
<td>625</td>
<td>560</td>
</tr>
</tbody>
</table>

Note: Performance for Cisco USP with either record route on or record route off will vary depending on call flows, which affect the SIP requests per second. Furthermore, maximum module performance will be lower when DNS lookup, SIP logging, or RADIUS logging services are enabled.

Table 12 shows Cisco Unified SIP Proxy Version 9.0 resource usage on a virtualized platform.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>100 SIP Requests per Second</th>
<th>200 SIP Requests per Second</th>
<th>300 SIP Requests per Second</th>
<th>400 SIP Requests per Second</th>
</tr>
</thead>
<tbody>
<tr>
<td>vCPU</td>
<td>2</td>
<td>4</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>RAM</td>
<td>4 GB</td>
<td>4 GB</td>
<td>4 GB</td>
<td>4 GB</td>
</tr>
<tr>
<td>Disk</td>
<td>80GB</td>
<td>80GB</td>
<td>80GB</td>
<td>80GB</td>
</tr>
</tbody>
</table>
Hardware Specifications

Cisco UCS hardware specifications for Cisco USP Version 9.0 and later can be found at the following URLs:

- [https://apps.cisco.com/ccw/cpc/guest/home.do#!/content/ucsSeriesDetails/series_cseries](https://apps.cisco.com/ccw/cpc/guest/home.do#!/content/ucsSeriesDetails/series_cseries)


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For More Information

For more information about Cisco Unified SIP Proxy, contact your local Cisco account representative.